

Amendments to the Claims

This listing of claims will replace all prior versions, and listings, of claims in the application.

1. (Currently amended) A portable voice over Internet Protocol (VoIP) test device for testing a VoIP network, comprising:

a user interface;

a transceiver configured to communicate with the VoIP network;

a memory storing a test algorithm;

a codec;

a media access controller (MAC); and

a processor in communication with said user interface, said transceiver, said codec, said media access controller, and said memory and configured to execute said test algorithm to cause said transceiver to communicate with the VoIP network to test at least one of the group: jitter, packet loss, and latency of the VoIP network.

2. (Previously presented) The VoIP test device of claim 1, further comprising a digital signal processor in communication with said processor.

3. (Currently amended) The VoIP test device of claim 2, wherein said digital signal processor ~~comprises at least one encoder/decoder forms said codec.~~

4. (Currently amended) The VoIP test device of claim 3 ~~1~~, wherein said encoder/decoder ~~codec~~ uses at least one of the following compression protocols: G.711a-law, G711μ-law, G.720, G.723.1, G.726, G.728, G.729, G.729A, and G.729AB2.

5. (Original) The VoIP test device of claim 1, wherein said transceiver comprises a power line modem for communication with a power line communication network.

6. (Previously presented) The VoIP test device of claim 1, wherein said transceiver comprises an Ethernet transceiver.

7. (Original) The VoIP test device of claim 1, wherein said transceiver comprises a cable modem.

8. (Original) The VoIP test device of claim 1, wherein said user interface device comprises an audio input device and an audio output device.

9. (Original) The VoIP test device of claim 1, wherein said transceiver comprises a digital subscriber line (DSL) modem.

10. (Original) The VoIP test device of claim 1, wherein said user interface comprises a manual input device and a display.

11. (Currently amended) The VoIP test device of claim 1, further comprising a media access controller wherein said processor is configured to execute said test algorithm to cause said transceiver to communicate with the VoIP network to test at least two of the group: jitter, packet loss, and latency of the VoIP network.

12. (Original) The VoIP test device of claim 10, further comprising a dual tone multi-frequency encoder in communication with said manual input device.

13. (Previously presented) The VoIP test device of claim 1, further comprising a communication interface port in communication with said processor.

14. (Previously presented) The VoIP test device of claim 13, wherein said communication interface port comprises a RJ-11 connector.

15. (Original) The VoIP test device of claim 13, wherein said communication interface port comprises a tip/ring interface.
16. (Original) The VoIP test device of claim 1, further comprising a Power over Ethernet module.
17. (Currently amended) The VoIP test device of claim 5, further comprising a wherein said media access controller forms part of said transceiver.
18. (Currently amended) The VoIP test device of claim 1, wherein the ~~digital signal processor uses at least one of the following data compression techniques: G.711a law, G.711μ law, G.720, G.723.1, G.726, G.728, G.729, G.729A, and G.729AB~~ 2 said processor is configured to execute said test algorithm to cause said transceiver to communicate with the VoIP network to test each of jitter, packet loss, and latency of the VoIP network.
19. (Original) The VoIP test device of claim 5, wherein the device receives power from a power line communication network.
20. (Original) The VoIP test device of claim 1, further comprising a network status indicator.
21. (Original) The VoIP test device of claim 20, wherein said network status indicator provides a mean opinion score (MOS) output.
22. (Previously presented) The VoIP test device of claim 1, wherein the device includes a handset and a base and said processor is disposed in said handset.
23. (Original) The VoIP test device of claim 1, wherein said processor is programmed to test the VoIP network based on at least one of the following: E-

Model, Perceptual Analysis Measurement System, Perceptual Evaluation of Speech Quality, Perceptual Speech Quality Measurement (PSQM), and PSQM+.

24. (Original) The VoIP test device of claim 1, wherein said memory includes an Internet Protocol (IP) address stored therein.

25. (Previously presented) The VoIP test device of claim 1, wherein said memory includes an algorithm for requesting an IP address stored therein.

26. (Original) The VoIP test device of claim 1, wherein said memory includes a MAC address stored therein.

27. Canceled.

28. (Currently amended) A method of using a portable test device to test a VoIP network, comprising:

transmitting test signals over the VoIP network;
receiving response signals in response to transmitting said test signals;
wherein the response signals are received from the VoIP network via a codec and a media access controller;
processing said response signals to determine the quality at least one of the group: jitter, packet loss, and latency of the VoIP network; and
presenting an indication of the quality a result of said processing of the VoIP network to the user.

29. (Original) The method of claim 28, wherein the processing comprises at least one of time-frequency mapping, frequency warping, intensity warping, loudness scaling, asymmetric masking, and cognitive modeling.

30. (Previously presented) The method of claim 28, wherein said presenting an indication comprises indicating at least one of the following: incorrect

Internet Protocol configuration, incorrect gateway address designation, signal echo, and call drop out.

31. (Original) The method of claim 28, further comprising determining whether the VoIP network is operable to communicate voice data according to predetermined voice communication parameters.

32. (Previously presented) The method of claim 28, wherein said processing comprises determining signal distortion.

33. (Currently amended) The method of claim 28, wherein said processing comprises determining each of, signal delay, jitter, and packet loss of the VoIP network.

34. (Currently amended) The method of claim 28, wherein said processing comprises determining at least two of the group: packet jitter, packet loss, and latency of the VoIP network.

35. (Original) The method of claim 28, wherein said indication comprises a MOS indication.

36-48 Canceled.

49. (Currently amended) A method of testing a VoIP network, comprising:
receiving an input from a user interface;
executing a test algorithm;
transmitting a first test signal over the VoIP network;
receiving a second signal from the VoIP network;
processing the received second signal via a codec and a media access controller; and

processing said second signal to determine a jitter, a packet loss, and a latency of the VoIP network.

50-55. Canceled.